20 Minute Project ...

Tunable Audio Filter

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Sometimes simple circuits can do wonderful things. This is certainly true of the simple active bandpass audio filter described here. Sure, we live in the age where things like DSP and switched capacitor chips give filter performance that nobody even dreamed of 20 years ago. This isn't a substitute for a high-tech filter. But with a dozen parts you probably already have in your junk box, you can build an audio filter that you'll probably like and that you'll use on the air. If you've never tried a simple audio filter, give this one a shot. If you've used an audio filter before, you already appreciate what they can do for you and you may want to try this one.

What can a simple audio filter do for you? It can help separate stations on a receiver that has no narrow filter. It can reduce lightning noise, direct conversion "hum", or circuit hiss. It can help you dig a weak signal out from the noise. This isn't a "brick wall" filter, and you'll still be able to hear all the signals, but you can emphasize the one you want. The Gaussian filter shape (rounded passband) does not ring and is not as fatiguing to listen to as some sharper filters. It sounds very much like the "Q multiplier" that was common in the 1960's.

The schematic for the basic filter is shown in Figure 1. This circuit is an old standard. It is a low to moderate Q bandpass filter. The performance (Q and upper frequency limit) is limited primarily by the gain of the opamp. It has been used in many radios by many people, from Ten Tec to my Pixie Deluxe. The basic filter does not require R1b, however adding R1b allows the high gain of the circuit to be tamed, and it provides a way to adjust the filter frequency.

What makes this filter different from the usual implementation is that by the addition of a pot for R1b it becomes a tunable audio filter. This is one of those "magic" serendipities that sometimes occur in electronics. Although the design equations for this filter are simple enough, there is a lot of interaction that occurs between component values. You can't change the capacitor value or either of the two resistors without changing the frequency, Q, and gain. But experiment-

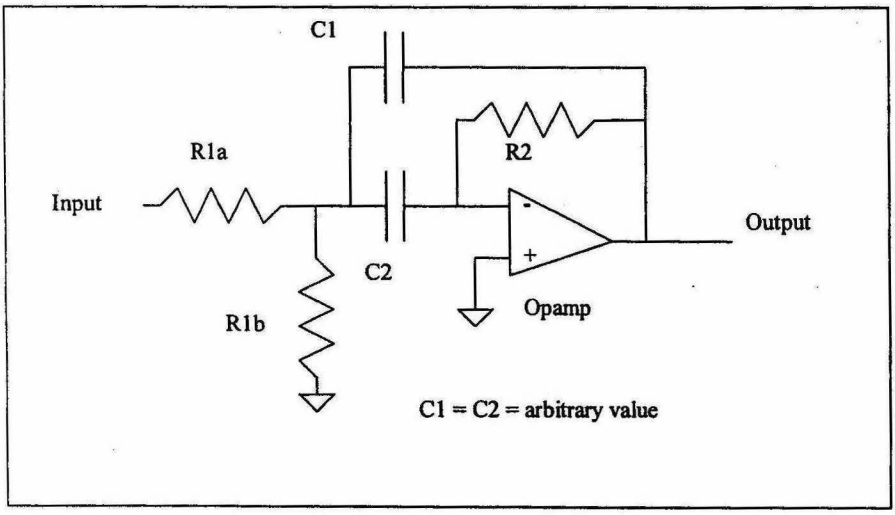

Figure 1 Basic Active Bandpass Filter Circuit

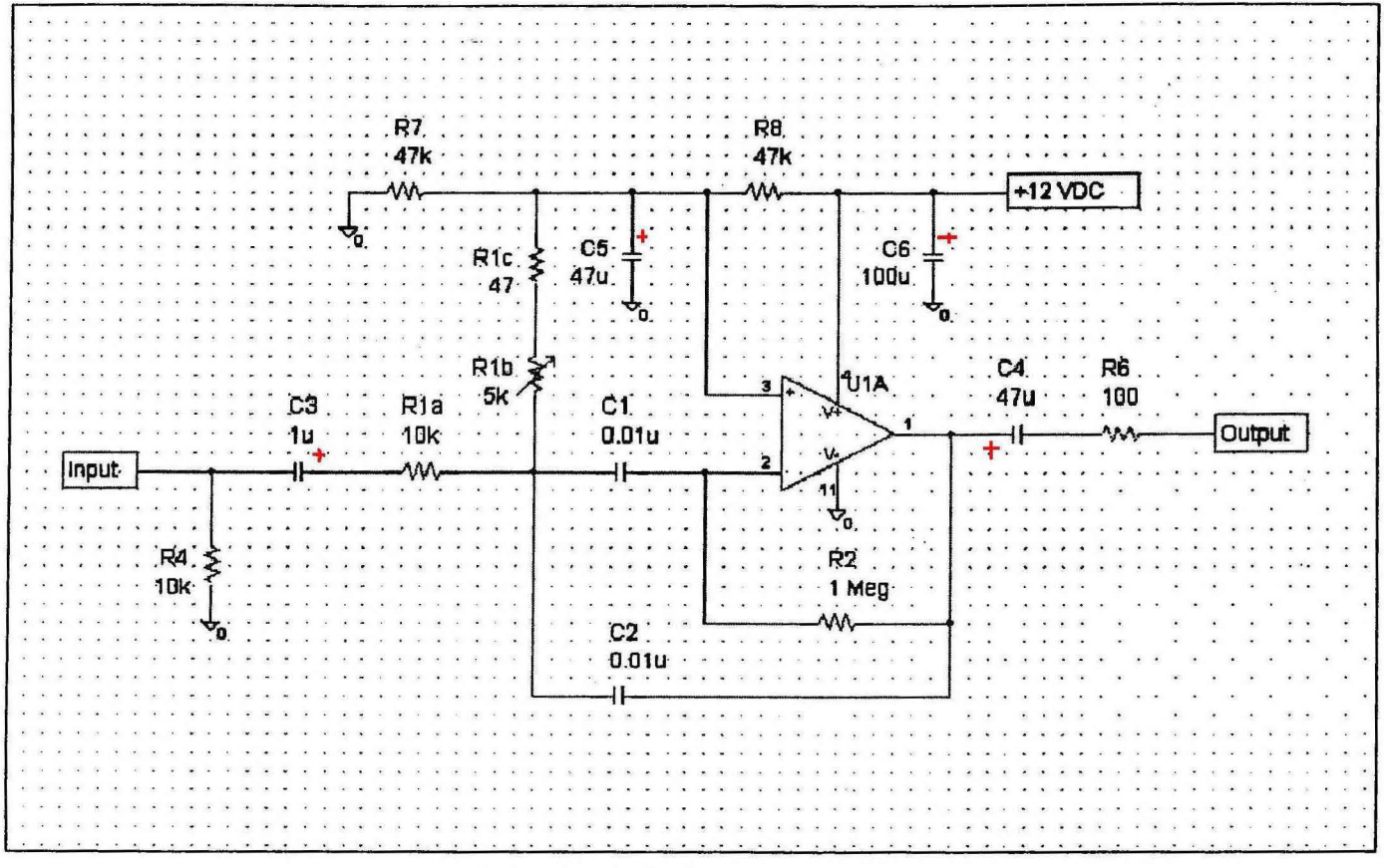

Figure 2 Complete Active Bandpass Filter Schematic

C1, C2 - non polarized (poly)

C3, C4, C5, C6 - electrolytic

ing with the circuit and some analysis shows some very interesting things. First off, using a pot for R1b allows you to adjust the peak frequency from very low to very high. The peak frequency is low with the pot at maximum resistance and high with the pot at minimum resistance. Secondly, the circuit acts as a constant bandwidth filter. As long as you don't run out of opamp gain, the filter Q increases as the frequency increases, with the result that the 3 dB bandwidth stays the same no matter what the frequency setting! And last but not least, the overall circuit gain also stays the same regardless of the frequency setting. These facts result in a very simple, user-friendly circuit.

By the time you add the other parts necessary for a practical circuit, the parts count has doubled. Figure 2 shows the real filter circuit. It gives a tuning range of about 250 to 2500 Hz and has some gain. The calculated 3 dB bandwidth is about 32 Hz. It will drive headphones to a good volume directly. The circuit was developed around a Rock Mite, which benefits substantially with a filter like this. Of course you could put this between the detector and volume control of an existing receiver, or add an LM386 audio power amplifier and drive a speaker with it. The source impedance must be much less than the 10k ohms of R1a or it will affect the filter frequency and Q.

There isn't anything really critical about the parts. The opamp should have a high bandwidth and open loop gain, because with high Q the circuit tends to run out of steam at high frequencies. The common TL-082 and NE5532 work fine, and there are many other types that should work. Some opamps have more bandwidth or output capability than others. The capacitors can be ceramic disks, polypropylene or polyester, or whatever you have. Since it is tunable, changing values only shifts your tuning range but doesn't keep it from working.

R4 is a load for the output of the previous stage. It should be on the order of 10 ohms if you're driving the filter from the headphone jack of a big receiver. Its necessary to provide a DC return on the Rock Mite so the muting MOSFET doesn't turn off, but the value can be high as shown. C3 is necessary to prevent the opamp bias from being drained by R4 or the external circuit. R1a, R1b, and R1c are part of the basic circuit. R1b is the filter tuning control, and R1c prevents the resistance from going too low and gives smoother tuning. R7, R8, and C5 provide a filtered bias voltage of ½ Vcc for the opamp. If the opamp were powered by +/- supply voltages, pin 3 and the top of R1c could be grounded directly. C1, C2, and R2 complete the basic filter circuit. C1 and C2 should be the same value. C4 is necessary to block DC voltage from the headphones, and I usually include the 47 or 100 ohm resistor (R6, value not critical and can be deleted) in circuits like this to provide a minimum load for the opamp; it also cuts down the headphone volume a bit. Of course, filtering on the power supply line (C6) is desirable, and a 0.1 u capacitor directly across the opamp power pins is a good idea.

The 8 pin opamps use pin 4 for the V- connection and pin 8 for V+. If industry standard parts like the TL-082 or NE5532 are used, the pin connections are the same and many different parts could be used or substituted for experiments.

Wire it up, connect headphones and an input, and power. The filter should be able to run off any power from about 9 volts to over 24 volts. The input signal should be kept low enough to avoid overloading the circuit; 50 to 100 millivolts is a good starting point. Depending on the opamp used, the circuit will drive 20 to 40 mA of audio and up to several volts into series connected walkman headphones (64 ohms). Of course, an LM386 audio power amplifier could be added to drive a speaker with good volume.

How does it work? Here's a simple explanation. C1 and R2 act together as a high pass filter. C2 and the R1's act together as a low pass filter. Put them together and you have a bandpass filter, with the opamp making up for the loss in the R-C filter sections. See the sidebar for further design information on this circuit configuration.

For how simple this project is, it is a very useful accessory to simple receivers. If you've never tried an audio filter before, toss this together on a breadboard and try it out. I think you'll be pleasantly surprised.

Sidebar:

Active Bandpass filter design equations and analytical data

For those who want more technical details, here are the design equations that are used to calculate component values and filter performance. This information is also available in many other texts. There is also a chart that show the relationship between component values and filter performance.

There are three basic relationships that define this filter. Given an arbitrary value for C1 and C2 (C1 = C2), and omitting R1b:

R2 = Q / (pi * F * C)

Where Q is the desired filter Q, F is the peak frequency in Hz, and C is the capacitor value in farads. Then:

R1 = R2 / (4 * Q^2)

And the circuit voltage gain is:

Av = 2 * Q^2.

There are also some derived formulas that are useful in analyzing the filter behavior:

Q = sqrt (R2 / (4 * R1))
F = Q / (pi * R2 * C)
Q = 1 / (4 * R1 * pi * F * C) = (R2 * pi * F * C)

These derived equations tell us that the Q is determined soley by the values of R2 and R1. It also tells us that, all other things being equal, the peak frequency is lower with larger values of C, as you probably already suspected. The picture becomes more complicated when R1 is split. The basic equations for R1a and R1b are:

R1a = R2 / (2 * Av)

R1b = (R2 / 2) / (2 * Q^2 – Av)

Where Av is the desired voltage gain of the circuit (which must be less than 2 * Q^2). For high Q's (Q > 10) and relatively low Av (Av < 20), an approximation can be made so that

Av = (R1b / R1a) * (2 * Q^2)

Which is obviously just the resistor divider ratio times the original circuit gain. And for frequency and Q calculations, the parallel combination of R1a and R1b are substituted into the design equations for R1.

As mentioned, all this holds so long as the opamp doesn't run out of gain. "But what!", you say, "My opamp is spec'd for a minimum gain of 100,000! Isn't that plenty of gain?" Don't forget that most all opamps these days have designed-in frequency compensation (i.e. the gain is on purpose rolled off at 6 dB peroctave starting at some low audio frequency so the part doesn't oscillate when you use it in a circuit!). The frequency-gain characteristic is usually specified as "gain-bandwidth product", stated as a frequency, and it is the frequency at which the gain drops to 1.0. Below that the gain increases by 6 dB per octave until it reaches its maximum value at very low frequencies. Given that, the gain at any specific frequency is the gbw divided by the frequency. So at 1000 Hz, your 1 MHz opamp only has a gain of 1,000,000/ 1000 = 1000 to begin with. And a filter Q of 31.6 requires an opamp gain exceeding 2000. As the opamp runs out of gain, the Q and circuit gain both drop. This isn't a problem however, and you won't notice any big change in performance with the filter as you exceed the required opamp gain.

Sometimes high powered circuit analysis programs are most useful for analyzing circuit behavior. When you have a few equations like the above, you can enter them into a spreadsheet and solve them for many different component values to get an idea of how the circuit acts when you change parts values. Some analysis was done in Excel, with the resulting data plotted in graphical form, to show the relationship between R1b, F, Q, filter bandwidth, and filter gain. See this interesting graph on the next page.